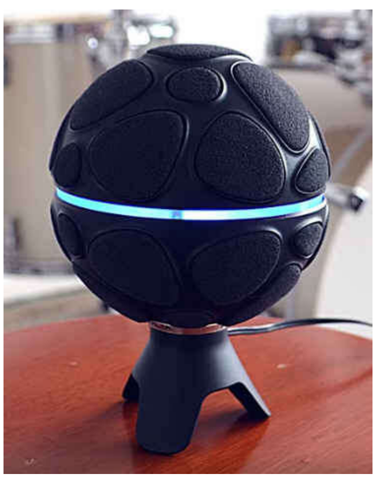
2017 Winter NAMM Show Report

Anaheim, California
January 19-22, 2017
Mike Rivers – ©2017

Here we are again, four days of cruising the aisles looking for new and cool stuff. The music industry is very much alive, loud, and busy. With over 100,000 attendees and dozens of first time exhibitors, there was plenty to see and hear. As usual, I'll remind you that the scope of this report is pretty limited. I write about things that I find interesting (not necessarily things I'm going to rush right out and buy) and I hope you'll find them interesting as well. Here we go . . .

Microphones, Pickups, and Preamps

There are always new mics and I can't cover them all. There were some new technologies and interesting approaches worth mentioning, so that's what I'll focus on here.



Zylia, a company out of Poland, showed a 19-capsule spherical microphone array that they designed for single-point multi-channel live recording. There are three parts to their system – the ZM-1 microphone itself, the Zylia Studio recording application, and Zylia Cloud for project storage and off-line signal processing. The microphone is about 4" in diameter, with a little tripod base threaded for attaching to a mic stand, or it can stand on a tabletop. Inside the sphere, in addition to the mic capsules, there's some processing, a 48 kHz, 24-bit A/D converter, and what comes out the USB port (which also powers the microphone assembly) is a composite signal. The Studio application (Mac, Windows and Linux) is, in essence, the system's DAW.

The workflow for a session goes sort of like this. Band members arrange themselves around the mic (which seems to work better than being in "stage" positions), and then record a bit into the Studio program. The program attempts to sort out what's where, and allows you to identify instruments and positions if the program doesn't get it right. Then you record a take and listen to the playback as a stereo mix. When you're happy with a take, you can shoot your recording up to the Cloud to have it separated into individual tracks. Much as I dislike the term, I have to say that there's some artificial intelligence going on here, or just magic.

One of the things you can do in the Cloud is to let a real engineer "up there" mix your tracks for you, which might be a good idea for bands that want to record but don't really know their way around a studio or a DAW. While Zilia has to do the job of separating the mic's output into individual tracks, once that's done, you can work on the mix, either using the fairly basic tools in the Studio application or export the tracks as WAV files and work in your preferred DAW.

They didn't actually have a band recording there at the show (it was too noisy anyway) but they had some pre-recorded examples that had been split into tracks to play with the Studio mixer. It was really hard to say from listening to the demos just how good the microphone is as a microphone, but I was impressed with the separation they were able to achieve given the setup of the band in a not particularly acoustically great room – and, no, I don't know where the vocal was coming from. In the demo, the keyboard player is singing and facing away from the mics, so I guess there must have been a PA speaker set up somewhere since the mic doesn't have any place for external inputs.

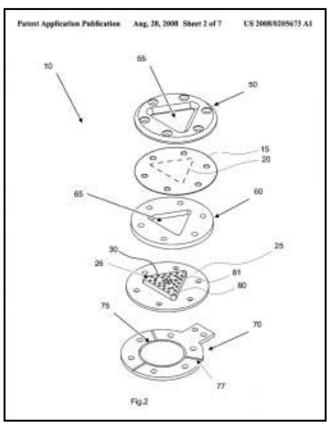
Regular price is $1299 for the whole shebang (nothing yet about costs associated with using the Cloud), but there's a pre-order deal going on now for $999 if you're interested. Give a listen to the demo files on the web site.
http://www.zylia.co/

Last year, Aston, a new British manufacturer, introduced two large diaphragm condenser mics, the Origin and the Spirit that are rapidly gaining a reputation as excellent sounding mics despite their modest cost. This year's new model, the Starlight, is a pencil-style medium diaphragm (about ¾") condenser mic with a gimmick – a built-in laser pointer that shows you where the mic is pointing. The laser beam shoots out from that hump extending behind the grille.

Now I would think that anyone who knew which end of a mic to blow into would know where it was pointing, but Aston's idea is that by having a laser-spotted point on a drum or perhaps a guitar body will help you put the mic in the same position the next time you mic that instrument. Honestly, I can't see that it would be that helpful unless you also knew the height of the microphone and angle of attack, neither of which you can get from a single laser point. A rep also suggested that when using a stereo pair, lasers could assure that they're both aimed at the same point, which is exactly what you *don't* want, unless you're fond of comb filtering. Also, some venues prohibit laser pointers.

But I digress. Otherwise it seems to be a well built mic that will probably find a lot of uses even if you never turn the laser on. The grill is made from a porous sintered material (thousands and thousands of tiny balls stuck together) which is nickel plated. There’s a group of switches along the stainless steel body that select attenuation (0, 10 or 20 dB), low-cut (flat, 80 Hz, 120 Hz), and a three position voicing switch. This gives you a choice of a modern condenser sound with a slight high end boost, a vintage sound with a bit of high end roll-off and a slight bass boost, and a hybrid combination of the two. Finally, there’s the on/off switch for the laser. A single mic with a standard clip runs about $350, and a stereo kit with two mics, two Rycote suspension mounts, and a stereo bar will also be available.
http://www.astonmics.com/starlight/

Ehrlund Microphones from Sweden is a new brand to me, though they’ve been in business for about 10 years now. What’s unique about these mics is that they have a triangular diaphragm. Pearl and Milab have been making mics with a rectangular diaphragm for many years, and the fairly new Audio Technica 5040 uses four rectangular diaphragms, so obviously a mic diaphragm doesn’t have to be round. Ehrlund’s capsule is actually built with a round diaphragm, presumably because it’s easier to tension properly, with a front mounting ring and drilled backplate giving it the triangular working area. This gives it two resonant frequencies rather than the single resonant frequency of a conventional round diaphragm, the advantage of which is that neither peak is very large, and it’s easier to smooth them out with the electronics. Frequency response is stated at a whopping 7 Hz to 87 kHz, but with no amplitude specs and no plots. They have several models, which appear to all use the same capsule and, other than sensitivity, have about the same specs. They differ primarily in case and grill construction, which tailors them to certain applications. At the top of the line, the is a large body cardioid condenser mic, with the EHR-T having two back-to-back capsules with separate outputs so that the pattern can be adjusted by mixing the two outputs – the new Lewitt LCT-640TS does this trick so we might be seeing more of it in the future. Other Ehrlund models include the stubby EHR-D for drums, the EHR-M1, a smaller and more robust version of the EHR-M that’s more appropriate for touring, and the EHR-E for high-SPL work such as drums and brass instruments.



http://ehrlund.se/

Unless you've been living on another planet for a month before the NAMM show, you've undoubtedly heard of the Slate Virtual Recording Studio. Steven Slate's video seems to be linked in just about every pro audio and recording forum. For the past few years, he's been talking about a virtual microphone system which finally began shipping last year. This is comprised of a large diaphragm condenser mic plus modeling software to emulate the sound of several iconic studio microphones. I've always taken a dim view of this because, while it's not difficult to model the on-axis frequency response of a cardioid microphone, what really makes a mic unique is how it behaves off axis. A mic that has fairly even response 30 degrees off axis, perhaps with the high end dropping off a bit, even the response of the best mics gets pretty wild and irregular, and that's what affects the room sound and leakage that's part of the mic's output. So the best you can do is emulate a good mic in a good room with minimal off axis leakage. Be that as it may, there seems to be some enthusiasm for having a virtual collection of well known mics that are frequently used for close-miked vocals in a good room, where the mic hears almost entirely the source you're recording. Slate presses on, though, in his quest to provide the capability of a great studio in a single package that you can carry home from your local music store.

New this year is the ML-2 small diaphragm microphone which models over 20 condenser and dynamic mics, and the VRS-8 Virtual Studio Interface. The VRS-8 includes 8 of the Slate VMS-1 mic preamps used with the original modeling microphone. The interface uses a new AKM A/D converter chip to a Thunderbolt computer connection. Latency is claimed to be 0.7 ms at 96 kHz sample rate, but it's not specified how it's measured. If that's the latency from the mic input through the A/D converter, the driver, the DAW, and back to the monitor output on the VRS-8, that's commendable. If it's the latency through the interface itself, which doesn't involve the computer, I've measured better than that with a Focusrite Scarlett interface. This is something I'm very fussy about, though I suspect I won't have an opportunity to test it myself.

I continue to remain skeptical. I don't think the Slate Virtual Recording Studio is going to put places like Blackbird Studio out of business, but what I see coming down the pike is that people who, in this century, are beginning to learn about recording will have a different set of tools to work with than those of us who were working in the 1970s, and will develop different techniques that take advantage of their tools just as we did with our tools. They'll learn something from the old timers – the mics they chose, but not why they chose them, and will follow suit. Watch the video, and don't let the applause get to you.
http://slatedigital.com/

Since the early 2000s, Chameleon Labs has been making good sounding analog mic preamps and other outboard gear at moderate cost featuring custom built transformer coupled inputs and outputs. A couple of years back, the company got a new owner who took the classic British designs that they'd been building and, while maintaining the sound that made them so desirable, brought them in line with the requirements of the modern digital studio such as lower noise, higher headroom, better EMI immunity, and form factor to suit less spacious control rooms. This show brought the first group of new products. The 7603 is a mic preamp and three band equalizer with a number of useful extra features. The mic preamp's 300/1200Ω switchable input impedance providing up to 70 dB of gain is complemented by a 100 kΩ instrument DI input and a balanced line level input. A low-cut filter at 40, 80, 160, and 320 Hz is followed by a 3-band equalizer with low and high frequency shelving at four frequencies each plus mid band peak/dip at six frequencies. There's a polarity reverse switch, a VU meter, in-house developed custom input and output transformers, internal power supply, and it's in a single rack space configuration. There's another model, the 7603-XMOD, which is identical except for the use of genuine British Carnhill transformers.

http://chameleonlabs.com

Focusrite added two new 8-channel mic preamps to their Scarlett series, the OctoPre and OctoPre Dynamic. These are intended both as stand-alone preamps and for input expansion for computer audio interfaces with ADAT optical input. Both offer eight mic preamps with phantom power switchable in two banks of eight, with XLR/TRS combo jacks for mic or line level inputs, two of which can be switched to high impedance DI inputs. There are eight five-step level meters on the front panel, eight analog line level outputs, A/D conversion up to 192 kHz sample rate with word clock in and out and dual ADAT optical outputs for 8 channels out up to 96 kHz sample rate, and 4 channels at 192 kHz. The Dynamic offers one-knob analog compression on each channel and ADAT optical input for D/A conversion, which can be routed to the analog outputs to provide additional headphone mixes or multi-channel surround monitoring. The Scarlett OctoPre has two combo input jacks on the front panel and the other six on the rear, while the Dynamic has all eight on the rear.

https://us.focusrite.com/adat/scarlett-octopre?reload=1#

IK Multimedia introduced the iRig Acoustic Stage, a preamp dedicated to last year's new iRig Acoustic clip-on guitar mic. The mic was initially designed for use in conjunction with an iOS mobile device running their Amplitube Acoustic processing app to fine-tune the instrument's tone. The Acoustic Stage replaces the app with a belt pack format box that offers three different tonal balance colors for steel or nylon string instruments, along with a Cancel Feedback button that switches in a set of notch filters that are likely to help. There's a line level analog output for connecting an amplifier or mixer plus a class compliant USB audio output for recording directly to a computer or mobile device. In addition, there's an auxiliary analog input for a guitar's built-in pickup that can be mixed with the iRig Acoustic mic. A polarity invert switch on this input is provided to get the mic and pickup closest to being in-phase. This could also serve as an input for pre-recorded backing tracks, or maybe even a vocal mic, depending on levels and gain.

http://www.ikmultimedia.com/products/irigacousticstage/

Last year I reported rather enthusiastically on the Tone Dexter from Audio Sprockets (see my 2016 NAMM report for a full description). Over the past year it got a bit of a makeover based on input from early testers and users. Briefly, it's a device that compares the sound of your acoustic guitar played into a good quality mic with what comes out of its pickup. Processing in the box does its stuff and matches the pickup sound with the mic sound pretty closely. The new version replaces some of the menu-selected items with dedicated switches, and they tweaked the algorithm a bit.
http://audiosprockets.com/

It might be stretching the point to call the Samson Go Mic Mobile a mic or preamp, but I thought it was pretty cute. It's a tiny 2.4 GHz wireless mic receiver that clamps on to the back of a mobile phone or to the flash shoe of a DSLR camera and outputs audio via a micro USB port or analog headset jack which, if not feeding a camera, can be used for monitoring. A USB-Lightning cable is included for connecting to an iPhone, and it should work with an Android device with USB-OTG support and running OS Version 5 or later. The dual channel receiver supports two simultaneous transmitters that can be mixed or recorded on individual channels. There are

three transmitters available, a handheld mic, a lavaliere with a belt pack, and a short shotgun.
http://www.samsontech.com/samson/products/wireless-systems/gomicmobile/gomicmobile/

Computer Recording Interfaces

Two major players in the lots-of-channels field, Antelope and Lynx each announced a new large interface with a Pro Tools HDX port, making them directly connectable to a Pro Tools HX system. This is a good thing, as it adds alternatives to Avid's own hardware, but it also means that the players will have to be ready to jump on to new driver development should Avid make a change in Pro Tools that requires it. It's a risk that's worth evaluating, though I expect it isn't going to stop many from being enthusiastic about the availability of alternatives.

The Antelope Orion32 HD offers HD connectivity via a built-in HDX port, plus native (and any other non-proprietary) connectivity via USB3. 32 analog channels in and out are on DB-25 connectors, with 24-bit A/D and D/A conversion at sample rates up to 192 kHz. A pair of ¼" TRS jacks provides an independent stereo output for monitoring. Digital I/O includes 64 channel MADI optical in and out, and just for good measure, 8-channel ADAT optical I/O. Antelope built their reputation on high stability, low jitter sample clocks (that was their first product, before they got into the interface business), and the Orion32 incorporates their latest clocking technology. There are two word clock or Loopsync (an Avid HD clock distribution system) outputs and inputs. A FPGA (Field Programmable Gate Array) provides firmware-based monitor mixing and modeling of a continually expanding library of signal processor emulation with low enough latency that it can be considered real-time (a small handful of samples, but not zero latency as advertised). A software application controls multiple output mixes as well as the modeled effects. The front panel is dominated by Antelope's signature sample rate display and a set of reasonably high resolution input and output level meters, plus five buttons to select preset setups and a couple of other functions that you'll probably need to read the manual once or twice to learn how to use them.

http://en.antelopeaudio.com/products/orion32-hd/

The Lynx Aurora has been a mainstay in multi-channel interfaces for quite a few years, and this year, Lynx introduced the Aurora$^{(n)}$ next generation interface. Along with the 8- and 16-challel stalwarts, 24- and 32-channel versions have been added to the line. The new series incorporates the converter technology used in their Hilo 2-channel interface for lower distortion and greater dynamic range that the previous series. L Slot expansion slots allow for a wide array of current connectivity options including Thunderbolt, Dante, ProTools|HD, USB (8- and 16-channel versions only) and a place to accommodate future formats. Other features include a micro-SD card slot for direct 32-channel recording and playback, two headphone jacks, and, coming later this year, future input and output expansion modules including mic preamps, analog summing, and AES3 and ADAT optical digital I/O.

http://www.lynxstudio.com/product_detail.asp?i=83

Universal Audio released the Apollo Twin MkII, a significant makeover of this popular and powerful desktop Thunderbolt-connected interface. Significant updates include new technology A/D and D/A converters for lower distortion and increased dynamic range, and incorporation of their Unison mic preamp modeling technology, which adds emulation of several classic mic preamps to the Apollo's mic and instrument DI inputs. In addition to the two front panel mic/line XLR-TRS combo jacks, eight additional inputs are available through an ADAT optical Toslink connector on the rear panel. The six outputs are counted as four TRS jacks on the rear and stereo headphone jacks on the front. Like all of the Apollo series, the Twin supports UA's hardware-based DSP plug-ins, and it's available with a single (Solo) two (Duo) or four (Quad) DSP cores. Other new features include a built-in talkback/slate microphone, more extensive monitor control with front panel Mute, Mono, Dim and ALT buttons, along with the ability to serve as an integrated desktop monitor controller if you have other Apollo units in your system.

http://www.uaudio.com/

Roland has a new line of compact desktop USB interfaces. The Rubix 22 is a basic 2 in x 2 out interface. The 24 adds two more line outputs plus a compressor on the inputs, and the 44 is like the 24 but with two more mic/line inputs. We don't have much in the way of spec yet other than that the converters are 24-bit, up to 192 kHz sample rate. One cool feature is that the signal present/clip LEDs have lenses that extend them into the top panel so you have a better chance of seeing if you're clipping when you're standing a few feet away at a microphone.

https://www.roland.com/global/products/rubix22/
(Replace 22 with 24 or 44 for the other models if they're not indexed on the web site when you get there.)

I'm going to cheat a little here. Mytek was at the NAMM show but I didn't see their booth. I did spend some time with them at CES, however, and got a lesson about high resolution audio streaming and saw their new D/A converter (the audiophiles call it a DAC for Digital to Analog Converter, pronounced "dack), the Manhattan II. Back before D/A converter chips took a big step forward half a dozen or so years ago, studios were buying high end D/A converters from folks like Benchmark so that they could hear more precisely what their A/D chain was doing to their audio. Audiophiles picked up on this, and when streaming audio became the preferred listening method and it became possible to download high resolution audio files, they started looking for the next best technology.

I'm not a high resolution fanatic myself (I just listen to "the radio" over the Internet), so it was at Mytek's booth at CES that I learned about MQA and how the on-line streaming service Tidal is starting to offer it. 24-bit 96 kHz audio is too much data to stream with real peoples' Internet connectivity, and even when compressed with a lossless encoder like FLAC, it's still a pretty big download. MQA (Master Quality Authenticated) is a new format developed by the Meridian Audio folks that offers high resolution audio streaming at about 1.2 megabits/second, something that most household Internet connections can handle. Mytek has licensed the MQA decoder from Meridian and incorporated it into their souped-up Manhattan II DAC.

The Manhattan II has a multitude of inputs – USB from a computer, S/PDIF coax and Toslink, AES/EBU, dedicated inputs for a DSD stream, three analog inputs, and an optional phono preamp, allowing it to serve as a monitoring hub from a variety of sources. There's a pair of analog outputs on XLR connectors, plus two headphone outputs from a high quality headphone amplifier. It's probably overkill for the average studio, but definitely worth consideration if you're doing

mastering, and in your spare time, you can listen to streaming music at higher quality than any of your friends.
https://mytekdigital.com/hifi/products/manhattan2/

Recorders

TASCAM introduced a couple of new rack mounted 2-track solid state recorders designed for production studios that have a need to transfer audio files from one physical format to another. The SS-CDR250N records to and plays from SD flash memory cards, USB drives, or CD/RW. Audio files can be copied from any medium to any other medium. It also has balanced and unbalanced analog, AES/EBU and coax S/PDIF digital inputs and outputs and records WAV files up to 24-bit 96 kHz sample rate as well as MP3 at various bit rates. A Dante card is available as an option. There's a dedicated hardware remote controller as well as an app for Android or iOS mobile devices. A useful feature when you're in panic mode is instant recording when powered on without having to press the Record Ready button.

The SS-R250N is quite similar to the CDR250N only without the CD drive. The CD-A580 has both a cassette and CD transport and allows dubbing cassettes to a USB drive as MP3 files. There are practically no details available yet so I'm not quite sure just what you can and can't dub from which to what. It's a single box that can take care of most of your consumer format dubs. There's some info on the solid state recorders on TASCAM's Europe web site, but noting about the cassette-equipped one yet.
http://www.tascam.eu/en/ss-r250n.html
http://www.tascam.eu/en/ss-cdr250n.html
http://tascam.com/ just for good measure

MikMe takes a little different approach to portable recording. The MikMe Microphone's form factor is block about 2-1/2" square and a bit under 1" thick, very nicely built (it's from Germany). It records 24-bit 44.1, 48, or 96 kHz PCM or MP4 files to 16 GB of internal memory. There's no real metering, but there's an LED to indicate that it's recording. It has an automatic level setting feature, but you can also adjust the record level manually. Their main bragging point is that it contains a 1" condenser mic capsule, but there's only one, so it's a mono recorder. There's a USB port for charging and file transfer, a mini headphone jack for playback and monitoring while recording, and there's an iOS app for remote control, and also records a Bluetooth stream from the MikMe in real time

for safety. The app also includes overdubbing, mixing and editing features. My first impression of it was that it isn't very practical, given that it's mono and isn't really designed to be whipped out of your pocket (sharp corners) and hand-held, but if it's a decent mic some may find it useful. It's an Indiegogo project and, I think, a little pricey - retail is $499, but the pre-production price is $399.
http://www.mikme.com

Mixers and Control Surfaces

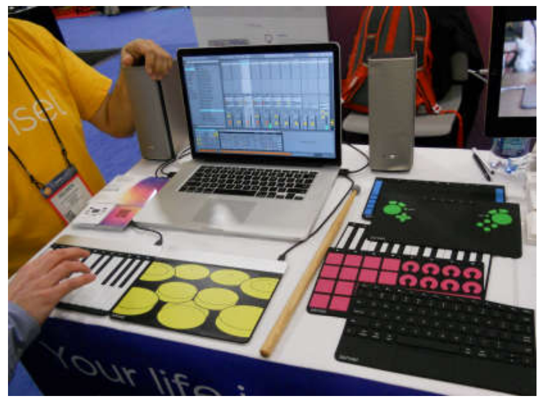
Let's start with control surfaces. Several years ago, PreSonus introduced the Faderport, a one-fader control surface for a DAW or whatever else needed it. It was good for riding a fader during a mix for those of us who mix more organically than by drawing a volume automation curve, but it worked with only one channel at a time. This year brings us the Faderport 8, an 8-fader, one big knob, one little knob, and 57-button desktop controller for a DAW. With modifiers (shift button) those 57 buttons control 78 different functions, far more than I can name offhand. It's kind of imposing, but the buttons are clearly labeled, several are user customizable, and once you get used to it, I suppose the location and function of the ones you use most of the time will become second nature.

The touch-sensitive faders are motorized so they'll follow automation written to a track, and when switching banks, they quickly and fairly quietly jump to match the DAW mixer's fader positions for the selected bank of tracks. A small LCD at the top of the "channel strip" displays the channel name, fader level in dB, pan position, and also an itty bitty level meter.

Each channel has a dedicated Mute and Solo button, plus a Select button. These match up with mouse clicking on your DAW screen, so pressing the Select button on a Faderport channel will do the same thing as selecting a track with the mouse. Tapping a Solo or Mute button latches the function until it's cleared while holding these buttons down restores the channel to normal when it's released. There are Select "modifiers" that make the channel Select button do something specific, for example, arming the track for recording. What may or may not be displayed on the LCE is dependent on the DAW with which you're using it. Obviously everything works as described when using it with PreSonus' Studio One program.

It offers Mackie Control Professional and HUI control protocols, which covers just about any DAW complicated enough to warrant a hands-on control surface.

There’s a special mode for Studio One that accesses some special functions that aren’t common to all DAW programs, and in an attempt to get you hooked, a copy of Studio One Artist is included. The manual has sections dedicated to other popular DAWs – Pro Tools of course, Sonar, Logic, Cubase/Nuendo, and Abelton Live. I’m hoping to snag one of these for review once they start shipping. The lack of hands-on control is what’s kept me from fully accepting a DAW in my life. Maybe this will change it.
http://www.presonus.com/products/FaderPort-8



While not exactly a control surface, the Sensel Morph could be one, or many, for different applications. It consists of a set of interchangeable pads that lay on a multi-touch pressure-sensing base about the size of a mouse pad. The pads are laid out for different functions – there was a drum pad, a sample player, a piano keyboard, a mixer, a blank drawing tablet, and a QWERTY keyboard. A strip on the back of each pad identifies its function and sets up the device’s output appropriately. Data output is from a USB port or Bluetooth transmitter, and it has an internal rechargeable battery for wireless operation. I’m not sure it would replace a Faderport 8 or something similar, but it might be great for controlling a plug-in. It’s a Kickstarter project, $249 for the base and one pad, $25 each for other pads.
http://www.sensel.com/

Just when I thought that the concept of a small or medium format modular analog console was dead, along came the Arthur Format 48 system from Schertler. This is a mixer frame and a series of channel input and summing modules that allow you to custom-assemble your own console. They offer two line/mic input modules, one standard and the other “ultra low noise,” a high input impedance instrument DI input module, a stereo line level input module, and two master modules. The frame is sectional, so you can assemble whatever size you need.

The mic/line modules have switchable 48v phantom power, a low cut filter, three auxiliary sends selectable pre or post fader, a three band equalizer with a bypass switch, sweepable mid band, and a variable 150 or 240 Hz notch that I suspect is intended as an anti-feedback measure since those seem to be pretty common frequencies for on-stage feedback. There's also a mute and PFL solo button, signal present, and peak indicators. There's a variable input gain control, a jack that can be either a direct output or channel insert at the press of a button, and of course a long through fader and pan. The DI module is similar to the mic/line modules, with a "warm" button replacing the low frequency notch filter.

The stereo line input module has a simple two band EQ, 3 auxiliary sends for each of the two channels, and, for convenience, both ¼" and RCA input jacks. Outputs are handled on two modules, one that has the left and right master faders, Aux 1 master, and a single effect return. The other output module has the Aux 2 and 3 outputs, headphone outputs and a talkback section with built-in mic

It's a "flat" mixer with all the connections on the top of the panel, and the modules look and feel very good. At present, it's only a stereo mixer with no subgroups, but they tell me that they're thinking about a subgroup module, which, unless there's some trickery with patching, would require a new input module with bus assignment switches. While it's not the answer to everyone's wishes, it does allow for quite a bit of customization, very much applicable to a musician who never uses more than two mics, but needs a lot of DIs or a big keyboard mixer. http://www.schertler.com/en_US/home

PreSonus has updated their StudioLive console. The new Series III follows the same theme of their original and 2nd generation AI series consoles, with the primary operational difference being that the row of knobs above the faders in the previous versions morphed into channel strip knobs for adjusting equalization and dynamics has been replaced by a touch-sensitive and larger LCD than on the older consoles that pops up when you select a channel, and that's where you make all your tweaks. Another new feature is expansion of their "Fat Channel" that now includes several component-modeled signal processing plug-ins to supplement the standard EQ and dynamics that they've had since the initial version. The Series III comes in two sizes, 16 and 32 channels, with the same feature set other than the number of faders and input channels.

As in previous versions, the console can be controlled remotely from a computer or iOS mobile device, and, as with the AI consoles, the touch-sensitive faders are motorized so when you recall a scene, the faders are in the correct position. The Series III offers more options for output groups. In addition to the main stereo bus and four dedicated subgroup buses, there are 16 FlexMixes that can be assigned as auxiliary, subgroup, or matrix mixes with dedicated outputs. For example, if you really only need two aux outputs for stage mixes, what would become unused output jacks on a more conventional mixer can be used for other purposes like a broadcast mix, a green room mix, or sends to the plate reverb up in the attic. You have a choice of displaying all of the input channels on one layer of faders or split the faders up so that some are inputs and some are outputs so as to keep what you work with most always at hand.

They've ditched Firewire in favor of AVB network connectivity, though there's still a USB port so the console can serve as the I/O to a DAW as well as streaming up to 55 channels over Ethernet. There's also a built in recorder to record all the input channels plus the stereo mix to an SD flash memory card. There are many new features here, and a lot of flexibility, so be prepared for some study, and exercise your creativity to make it into the very special mixer that you want.
http://www.presonus.com/products/StudioLive-32
http://www.presonus.com/products/StudioLive-16

Signal Processors and Effect Boxes

Chameleon Labs discontinued their 7720 stereo compressor a couple of years ago. This year they showed an updated version that, according to the brochure I picked up at the show (and reported as a daily show tidbit), was also called the 7720. That's confusing.

Taking a second look at the brochure and getting out my jeweler's loupe, I see that the photo says 7730, so let's say that the new model is indeed the 7730. It's a VCA compressor like the 7720, and here's a rundown on the new features. It now has two VU meters (left and right channel) rather than one, it offers a choice of peak or average level detection, the onset of gain reduction can be set to either soft or hard knee, and there's a blend control to sum the input signal with the compressor output to achieve what's popularly called "parallel compression." Oh, and the front panel is now black rather than white.
http://www.chameleonlabs.com/

I've reported on the Softube Console 1 previously, so I won't go into too much detail here about this year's Mk II. To refresh your memory, Console 1 is a hardware control surface for DAW plug-ins – not every plug-in, but Softube has an extensive library that includes a full SSL 4000E channel strip, SSL 9000K, and "British Class A" with more in the works. New with this model is the ability to control Universal Audio's UAD-2 DSP plug-ins as long as you have the UA hardware (like the Apollo series) that supports them. While the basic controller is compatible with all of the major DAWs, it's more closely integrated with certain

DAW programs, allowing you, for example, to use buttons and knobs on the Console 1 to select tracks, adjust volume, plug-in send level, panning, solo, and mute. Presently this feature is implemented for Sonar and Studio One, with more on the way. There have been some minor appearance changes to improve legend and indicator visibility, and most important, as a result of moving principal manufacturing from Sweden to China, the price dropped in half, to $499.
https://www.softube.com/console1.php

The Sonicsmith ConVerter and Squaver P1 lie somewhere in between a signal processor and a musical instrument, so these units make a good transition between studio signal processing and a couple of stomp boxes that I wanted to write about. Both are based on their Audio Controlled Oscillator (ACO) chip that takes in a monophonic analog audio signal, analyzes its fundamental pitch, and puts out a square and a sawtooth wave. These waveforms feed a set of analog synthesizer processing circuits, and what comes out sounds like, and can be controlled like an analog synthesizer. The source can be an electric guitar, a keyboard (from an analog output, of course), even a microphone so you can play the synthesizer with your voice or your trumpet. Oh, and they accept MIDI as a control source for variable parameters.

I apologize for not being sufficiently conversant with analog synth technology to confidently give you a more detailed explanation without sounding like I'm babbling from the marketing materials, so I'll tell you what I learned that I remember. The Squaver is the more advanced one of the pair that offers more control over the synthesized waveforms, the ConVerter is the basic model that, by itself, doesn't give you a lot more to work with than a blend between the square and sawtooth waves and apply an envelope follower. The Squaver offers a voltage controlled filter and some other modulation possibilities.
http://sonicsmith.com

A few years back, TC Electronic released a set of effect pedals that were for specific processes, but that had downloadable settings for various parameters for which there were no external controls. Initially, the settings were made according to the wishes of some famous guitarists, and eventually they released an application so that you could edit them yourself. Well, ModDevices took this concept a giant step forward with the Mod Duo. It's an innocent looking stomp box with two knobs, two pushbuttons, two LCD screens, two inputs, and two outputs. With all that, you can create a monster pedalboard, or a simple one, if you prefer.

They offer an on-line library of virtual effect pedals that you can download and, via a USB connection, store your chosen effects in the Duo box. Then the fun starts. Using their application, you drag pedals on to the "board" and connect them together in whatever order you choose. In addition to cables, there are also splitters for creating parallel processing paths, and combiners to send parallel outputs paths to one output jack.

When it comes to playing, the buttons can switch one effect at a time in or out, or switch between different pedalboard configurations. The knobs allow you to adjust parameters of the selected effect. As you might imagine, it can get pretty deep, but even I, who has never used a stomp box (I'm an acoustic player that likes it that way), was pretty impressed. At present, all of the Mod Devices plug-ins are free, though they plan to license and model some commercial pedals, which will likely not be free. Or you can create your own plug-ins with MaxDSP. https://moddevices.com

The press release from Denmark's T-Rex said they were introducing a real analog tape based effect device that was similar to the Binson Echorec, and indeed they did. On my first look around the tape echo Replicator, I noticed a sticker on the back that said that the reels might not rotate during operation, and that this was normal. My first thought was "Great googly moogly! The tape mechanism is for show and that it was all done with electronics," but, no, it's a real, functional tape loop recorder/player. The tape is in a removable cartridge assembly that includes all the guides and pressure pads. The "reels" serve as part of the tape loop guiding system and don't store tape, so, in a sense they really are partly show, though it's an effective design for their job – to support the slack part of the tape loop. The tape loop itself is under pretty low tension and depends on pressure pads against the heads for good contact and wrap.

There's an erase head, a single record head, and two playback heads that can be switched to either provide two different delay times, or one can be switched to feed back to the record path to create repeating echoes. Delay is adjusted by varying the tape speed, which suggests that long delays might suffer a little loss of high frequency

response, however the electronic motor speed control allows for setting delay to match tempo by tapping a button. Speed can also be controlled from an externally applied control voltage. As expected today from any device that relates to analog recording tape, there's a Saturation control. Although it uses cassette width tape, it appears to be nicely built and, with reasonable maintenance, should work well. The Replicator is available in two formats, a stomp box and as a Eurorack module to scuzz up your analog synths.
http://www.t-rex-effects.com/replicator

Last, and certainly not least, though here only because I like nutty inventions that actually work and I admire machinery that's both functional and beautiful, I present the Zvex Candela Vibrophase. The boring description is that it's a phase shift effect device, but the interesting part is how it gets there. Working backward toward the middle of the contraption, there's an electronic package where the phase shifting takes place. It's controlled by the output of a photocell, which is modulated by a rotating disk with a mask pattern printed on it that varies the intensity of the light falling on the photocell.

So we've got our phase shift network, and now we have to rotate the disk to vary the amount of phase shift. The disk is driven by a Stirling cycle piston engine through a series of rods, cranks, and an eddy current brake controls its speed of rotation. A tea candle heats air in a chamber, which expands to drive the pistons, and it's recycled as it cools down to push the pistons through another cycle. The candle also provides light for the photocell modulator, and illuminates a solar cell that powers the electronics. Professor Lucifer Butts would be proud. Watch the video.
http://www.zvex.com/about-the-candela-vibrophase
https://www.youtube.com/watch?v=3gwZ2lHtG0Y
http://auto.howstuffworks.com/stirling-engine1.htm

Speakers and Other Monitoring Stuff

20 years or more back, Aspen Pittman, the father of Groove Tubes, together with engineer Drew Daniels, developed a single box stereo speaker that uses the M-S principle. Inside the box, there's one speaker that fires forward and another firing sideways. Left and right channels of a stereo audio signal are processed through a sum and difference matrix and sent to the forward and sideways speakers respectively. When they mix in air, the left/right stereo image is reconstructed. The concept never really got a lot of traction, though Fender made an amplifier using the technology, and Pittman himself brought out the compact Spacestation stereo box. At this show, he introduced a new and larger model, the CPS (Center Point Stereo) Spacestation XL. The XL uses a two-way bi-amped center speaker system with a 12" woofer and 1" titanium dome tweeter that reproduces the left+right sum. The side signal comes from two side-firing 6.5" drivers wired in opposite polarity so that a positive-going side signal will blow air out one side and a negative-going side signal will blow air out the other side. The center and side speakers are time-aligned so that the sound from all three speakers appears to be radiating from a single point in space. There are two independent stereo inputs, each with a level control so they can be mixed to taste. Another front panel control adjusts the stereo width by varying the relative level between the front and side speakers. Dimensions are 29" x 17" x 16", weight is a hefty 65 pounds.
http://aspenpittmandesigns.com/cps-spacestation-xl/

Moon Amplification showed a rotary speaker system (think "Leslie") with no moving parts except for the voice coils of the loudspeakers inside the box. There's some DSP voodoo controlling the phase of the speakers to simulate a mechanically rotating speaker. The processor produces the apparent rotary motion by continually changing the phase relationship between the speakers.

The top cabinet has four horns, one on each side, with ports on each corner. Each horn is powered by its own 250 watt amplifier. This is paired with a bass rotary cabinet with eight 8" drivers with 250 watts powering each pair. The photo shows the complement of speakers in the top and bottom cabinets. For extra bass power, there's a non-rotary sub-bass

cabinet with two 15" speakers, each driven by a 500 watt amplifier. There are a number of configurations, some rugged enough to get tossed in the truck and hauled to a different gig every night, some pretty enough to go in a church. It's a pretty cool idea, and as far as I could tell in the noisy hall, it works.
http://moonamp.com/index.html

Little Labs, the problem-solver expert, has a new little problem solver. The Monotor is Jonathan Little's take on what's become a popular product, the headphone amplifier. His products have a reputation for doing their intended job well, and he makes them unique in the field by incorporating some useful features you never thought you needed.

In addition to driving plenty of headphones plenty loud and clear, the Monotor offers switching for mono (left + right), left/right reverse, left or right channel only to both ears, and left minus right for a quick listen for out-of-phase material or hearing one of the detrimental effects of MP3 data compression. The volume pot can be bypassed if you have a D/A converter with a very good volume control and want to avoid one more component in the signal path. But this is Little Labs; there are two switches for this, one for the left channel and one for the right. By bypassing one channel and switching to the Mono mode, you can feed a mix to one channel and "more me" to the other channel and use the volume control to adjust the amount of "me" in the headphone mix. There two headphone outputs, each with both a ¼" and mini phone jacks. Each pair of jacks, though sharing a single volume control, is driven from a separate amplifier so that plugging in the second set of phones doesn't affect the level of the first set, though each of the output pairs can adequately drive two sets of headphones of any of today's wide range of impedances. Finally, there's a pair of ¼" TRS jacks in parallel with the two combo XLR input connectors for a pass-through to your monitor speakers. Interestingly, everything in the box is passive up to the amplifiers that drive the headphones. It's solidly built, and it's 1 rack space high and ¼ of a rack panel width, so four can be mounted together for a great studio headphone system.
http://littlelabs.com/index.html

Back in 2001, I invented the Monitor Controller, or at least I think that maybe I did. In the February 2001 issue of Recording Magazine, I wrote an article about planning and building do-it-yourself problem solvers. The problem I used as an illustration was one that was pretty common in the dawn of the DAW days – "How do I hear my monitors since I sold my mixer?" – and the solution was a monitor controller, actually three, at different levels of complexity. A couple of years later, I started to see commercial products appear that did the job. I hope somebody built one from my article before buying one off the shelf, but I never found out.

One the popular commercial monitor controllers from the early 2000s was the Mackie Big Knob. It was a very good design, but it eventually faded into the sunset, I suspect, when DAW users graduated from simple sound cards to outboard computer audio interfaces. With enough panel space on the interface box for a few switches and knobs (and later, with the evolution of the software control panel), monitoring became an integral part of the interface. Once the users figured out how to use the monitoring facilities in the interface (it took a while), there was no longer a need for a separate controller. But what went around comes around again, especially if it's a good idea and well implemented.

At this show, Mackie re-introduced the Big Knob monitor controller, in three versions this time around. The Passive is a true passive pot in a box offering a selection of two stereo input sources and two stereo outputs, mute and dim switches, and a big volume control knob. The Studio and Studio Plus offer 3 inputs and two outputs and 4 inputs and 3 outputs respectively with independent trims on all of the inputs and outputs.

The Studio models include input level meters, a USB recording interface (24-bit, up to 96 kHz sample rate on the Studio, 192 kHz on the Plus) with two Onyx mic preamps, an input/output mix control for true no-latency monitoring when recording, and two headphone outputs with independent volume controls. The Studio has a built-in talkback mic; the Plus has a connector for an outboard talkback mic as well as the built-in one. One feature included on the original Big Knob that's missing with the new ones is an RIAA-equalized phono input, particularly with the rising interest in phonograph records, but I guess you can't have everything.
http://mackie.com/products/big-knob-series

I wasn't sure if I should leave the Half One loudspeaker for the crackpot ideas section of this review, but since there was some impressive research behind it (and I still think it's a crackpot idea) I decided to put it along with other monitor-related items. It's a loudspeaker that's been cut in half through the cone and frame, but with the voice coil left intact. At first look I thought it was a

loudspeaker manufacturer showing their insides, but no, it's actually supposed to play like that, and in free air (no box or baffle) to boot! It comes from Onkyo, a perfectly reputable audio equipment manufacturer, who engaged in an engineering study with the goal in mind to make a loudspeaker that has a radiation pattern like that of a musical instrument, though they don't say what instrument. Their 3D polar plot shows several lobes and nulls which is indeed how instruments radiate, so I guess they achieved their primary goal, at least in the laboratory. However, they sounded pretty horrid as might be expected by looking at the peaks and nulls in the anechoic chamber frequency response plot at any measuring angle, even on axis. They're priced at $3,000 each, $5,000 for a pair. If you love specifications, visit the web site, really.
http://www.nihon-onkyo.co.jp/technical/contents.php?id=12

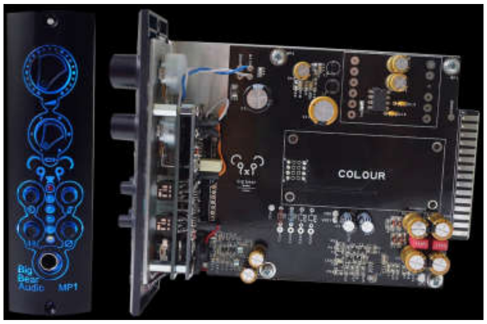
The 500 Club

There were the usual gaggle of signal processors and mic preamps in the 500-series module format, but nothing that really jumped out at me. But since I spent some time at the Chameleon booth and learned that they jumped on that bandwagon, here's a bit about what they're doing.

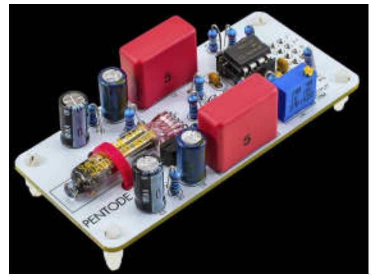
Chameleon introduced their model 880 8-slot frame with XLR inputs and outputs for every slot, a pass-through switch to feed the output of one slot to the input of the next slot without requiring a cable. DB-25 connectors parallel the XLRs. It comes with both a set of rack mounting ears and, for tabletop use, a removable top handle and rubber feet. And knowing that not everyone has eight modules ready to pop in, it comes with four blank panels so your one or two modules won't look so lonely.

And that one lonely module, Chameleon has the half-rack width, single rack space high CPS503. A module fits in the housing horizontally, and two can be put together with a little coupler for a single rack space 2-module rack. It's powered by a line lump supply that provides both the analog rail voltages and 48v for phantom powering. One power supply can power up to six chassis, so after the first one, there's a cost saving.

To keep those racks warm, Chameleon offers the 560 equalizer, the equalizer section of the 7703 in a 500 series module.
http://chameleonlabs.com



Big Bear is a new British company that showed their MP1, a 500-series mic preamp that's not a clone of anything, but was designed to be as clean and transparent as they could make it. But knowing that many people want some color to their mic preamps, they hooked up with an outfit called DIY Recording Equipment, who makes a line of Colour modules. These are small analog circuit boards that add various flavors of distortion color to an audio signal, and share a common physical format and pinout for power, input, and output. There's a blank space on the Big Bear MP1's circuit board where any of the DIY plug-in modules can be dropped in, or the Colour module can be bypassed for a clean preamp. The Colours run the gambit of tape saturation, transformers, a pentode, cassette distortion, and some others, most of which are $50 or less. I've run across a couple of mic preamps that have space and connectors for plug-in transformers, but these Colour modules take that concept up a level. Big Bear plans to design some Colour format modules of their own.



The Big Bear MP1+ is identical to the MP1 but with the MP1's electronically balanced output stage (seen in the upper right corner of the photo above) replace by a Lundahl transformer.
http://bigbearaudio.com/
https://www.diyrecordingequipment.com/collections/colour

Workstation Computers

It's nearly impossible these days to work with audio and not have a computer involved in some phase of the project. What's a novice to do who wants to start recording and producing music but doesn't know anything about computers beyond how to spell "Facebook?" Common advice is from experienced users and experts in the field is:

1. Keep your music production computer separate from your everyday computer
2. Don't buy an off-the-shelf computer, assemble it yourself
3. Optimize the operating system for running audio applications

That's all sound advice (note that I didn't include "Get a Mac" because that's outdated sound advice), though while #1 doesn't require much knowledge, only money and space, there aren't many people who, at least the first time, can be successful at #2 and #3. There have been a few shops that specialize in assembling and optimizing a computer for music applications – Sweetwater Sound is one, PC Audio Labs is another – but you have to know to look for them.

Now TASCAM has joined the club, and this is important because they're one of the companies that beginners (as well as experienced users, but I'm not talking about them here) look to for their gear. Further, they tend to look in places where TASCAM products have a strong presence – local music stores and on-line retailers.

TASCAM has long been a source for no-hassle integrated recording and mixing systems from the first cassette-based Portastudios to their current line of one-box digital production workstations. At this show they introduced Track Factory, which, while deviating from the one-box solution, offers the buyer a complete system-engineered set of components comprising a turnkey computer-based music production system.

The Track Factory bundle includes a TASCAM US-2x2 interface, along with an Intel NUC (Next Unit Computing) computer, plus a TASCAM TM-80 mic and TH-02 headphones. The computer is equipped with a 2.7 GHz i5 CPU, 8 GB of RAM, a 256 GB solid state disk drive, 4 USB ports, a Gigabit Ethernet port, and mini Display Port and mini HDMI video ports. Running Windows 10, it's hearty enough to support just about any of today's and likely tomorrow's music production software. Cakewalk's Sonar Professional is pre-installed and the operating system is optimized for audio by PC Audio Labs, a well respected supplier of turnkey music computers.

The package includes a keyboard and mouse, but you'll need monitor speakers and a video monitor in order to complete the system. Target price is $1300, which, when adding up prices of all the pieces from my local Micro Center and Guitar Center stores, means that you're paying about $200 for the integration and packaging, but for that you don't just get a ready-to-play recording system, but one that's expandable as your needs grow.

It was interesting to learn why TASCAM decided to go this route. It turns out that a substantial number of their lower end interfaces get returned because the buyers, usually with no more computer experience than knowing where the power switch is, have difficulty getting up and running, and that many dealers no longer have the expertise to help them out. Or maybe they don't think to seek help from their dealers, go on the Internet, and get flummoxed. By providing a turnkey system, a beginning user can get to work and by the time he's ready for

an 8-channel interface or a new plug-in, he'll have learned enough about his computer to make his next step up more comfortable and rewarding.
http://tascam.com/news/display/12606/

Stuff That Doesn't Fit Anywhere Else

There's a new source for audio transformers from Canadian manufacturer, Hammond Manufacturing (no relation to the organ company). They've been a regular supplier of power transformers, being the first place I look when a guitar amplifier needs a new power transformer (one of those was an exact replacement for the dead power transformer in my Hewlett-Packard 201C oscillator), and now they've added audio transformers to their product line. Not necessarily better or worse that the standards like Jensen and Lundahl, Hammond offers mic input, mic splitter, line input, line output, and a direct box transformer, with or without Mu metal shielding and with common form factors and sizes. Also, lest you don't already know, they also have a very full line of project boxes, should you be inclined to build something.

http://www.hammondmfg.com/

Jocavi is a manufacturer of acoustic treatment materials that's new to me. They make the usual diffusers and absorbers in their own shape and sculpture design, but what caught my eye was their Abstract line of membrane absorbers with the low frequency diaphragm absorber part being tunable between 50 and 250 Hz. There's an inflatable bladder behind the panel that adjusts its stiffness, and hence its low frequency absorption peak. There are curves on the web site that show that the low frequency absorption peak actually does move with changes in pressure, not very far, but enough so you can hit what's likely your most troublesome room mode right on the money. It's easier than moving a wall.
http://www.jocaviacousticpanels.com/uk/products/tun.abstract/index.htm

Tempo Technologies makes inventory tracking systems that use RFID stickers. This is kind of a business thing, but they recognize that it has applications for musicians or touring groups, which is what brought them to NAMM. For example, when packing the truck, you can scan each box to be sure everything's loaded and know to whom it belongs. One thing that I didn't know is that Android phones have the ability to read RFID tags, so you don't necessarily need a dedicated scanner. A check on the Google Play Store confirmed that there were indeed apps for scanning RFID tags, and that the apps are not compatible with either my phone or tablet, but I'm getting used to seeing that. Anyone want to donate your last year's Android to a good cause?

As part of their MusicLife program, Tempo Technologies maintains a “Find Me” database that could be really useful to musicians. If you have an RFID sticker on your instrument and it’s registered in their database, if you report a stolen instrument, there’s some chance that it could be found. Where? Well, apparently pawn shops are beginning to use the technology and scan for it when they’re offered something that might have an RFID tag on it. Anything that helps recover stolen property is good.
http://www.tempotech.com/musiclife.html

Roland introduced a new stage piano, the RD-2000, which has more controls on it than any piano deserves, but I’ll leave a detailed description to those who can do better than I on this subject. But there’s something that I can talk about that’s pretty interesting, both as technology and a little woe and intrigue. When connected to a computer, it transmits an enhanced MIDI stream that provides greater resolution of performance parameters than is part of the current MIDI specification. One of its enhancements is multiple polyphonic expression, for example, pitch modulation of individual notes in a chord.

In order to make use of this feature, you need sounds that understand it, and since nobody had any, Roland created some sound libraries and tucked them away in The Cloud. You can play something on the RD-2000 using conventional sounds, then upload the high resolution track to the Roland Cloud library, pick a sound you want, and your expression-enhanced track will be returned to you, rendered as an audio file that you can import into your DAW project to replace the original MIDI track. Roland Cloud is a subscription service (first month’s free, at least until the official launch later this year) that promises access to sounds from many Roland synthesizers.

A few years ago, a number of MIDI manufacturers, including Roland, wanted to explore incorporating Multidimensional Polyphonic Expression (MPE) into the MIDI specification. The MIDI Manufacturers Association (MMA) encouraged them to jointly develop, adopt, and publish a standard, but then Roland didn’t want to play. And then – surprise! They came up with their own system. MIDI was developed jointly by the early synthesizer manufacturers as a non-proprietary standard, and 34 years later, it continues to evolve. Maybe Roland’s MPE implementation will be incorporated into the MIDI specification, maybe not. MMA is hoping to be able to publish MPE extensions to the standard later this year.
https://www.roland.com/global/products/rd-2000/
https://www.rolandcloud.com/
https://www.midi.org

The Wrap

Talking about synthesizers seems like a good place to draw this report to a close with a mention of the growing interest in analog synthesis. As in the past several years, there's been an analog synth "island" that continues to bring new products, but highly sophisticated ones with huge racks and patchbays along with little tabletop boxes that let you be creative with waveforms, filters, and modulators for fifty bucks or less. Moog's piece of the analog synth area this year was a little booklet as a tribute to the passing of several well known developers, composers and artists who have contributed to the art and craft. Their theme was "Legends never die, they multiply." There was a listening station, of course, but not a fancy digital playlist . . . no, not for this crowd. They had a pile of cassette players loaded with music of the celebrated legends.